120 AF #30

Docket No.: W1878.0109/P109

8/25/04

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Patent Application of:
Atsushi Sasaki

Application No.: 09/039,072 | Confirmation No.: 9140

Filed: March 13, 1998 | Art Unit: 2644

For: LOUDSPEAKER UNIT ADAPTED TO ENVIRONMENT | Examiner: L. A. Grier

APPELLANT'S BRIEF

U.S. Patent and Trademark Office
220 20th Street S.
Customer Window, Mail Stop Appeal Brief - Patents
Crystal Plaza Two, Lobby, Room 1B03
Arlington, VA 22202

Dear Sir:

This brief is in furtherance of the Notice of Appeal, filed in this case on May 28, 2004.

You are hereby authorized to charge our credit card for the fee of $330.00 required under Section 1.17(f). PTO Form 2038 is attached.

In the event a fee is required or if any additional fee during the prosecution of this application is not paid, the Patent Office is authorized to charge any underpayment or credit any overpayment to Deposit Account No. 50-2215.

07/29/2004 JBALINAN 00000014 09039072
01 FC:1402 330.00 OP

CONTINGENT EXTENSION REQUEST

If this communication is filed after the shortened statutory time period had elapsed and no separate Petition is enclosed, the Commissioner of Patents and Trademarks is petitioned, under 37 CFR 1.136(a), to extend the time for filing a response to the outstanding Office Action by the number of months which will avoid abandonment under 37 CFR 1.135. The fee under 37 CFR 1.17 should be charged to our Deposit Account No. 50-2215.

This brief is transmitted in triplicate.

This brief contains items under the following headings as required by 37 C.F.R. § 1.192 and M.P.E.P. § 1206:

I. Real Party In Interest
II Related Appeals and Interferences
III. Status of Claims
IV. Status of Amendments
V. Summary of Invention
VI. Issues
VII. Grouping of Claims
VIII. Arguments
IX. Claims Involved in the Appeal
Appendix A Claims

I. REAL PARTY IN INTEREST

The real party in interest for this appeal is:

NEC Corp.

II. RELATED APPEALS AND INTERFERENCES

There are no other appeals or interferences which will directly affect or be directly affected by or have a bearing on the Board's decision in this appeal.

III. STATUS OF CLAIMS

A. Total Number of Claims in Application

There are 10 claims pending in application.

B. Current Status of Claims

1. Claims canceled: None
2. Claims withdrawn from consideration but not canceled: None
3. Claims pending: 1-10.
4. Claims allowed: None.
5. Claims rejected: 1, 4, 6, 8, and 10.
6. Claims objected to: 2-3, 5, 7, and 9.

C. Claims On Appeal

The claims on appeal are rejected claims 1, 4, 6, 8, and 10 and objected to claims 2, 3, 5, 7, and 9.

IV. STATUS OF AMENDMENTS

Applicant filed an Amendment After Final Rejection on April 15, 2004. The Examiner responded to the Amendment After Final Rejection in an Advisory Action mailed May 7, 2004. In the Advisory Action, the Examiner indicated that Applicant's After Final Request for Reconsideration was considered but deemed not to place the Application in condition for allowance. The claims in Appendix A incorporate all of Applicant's amendments filed prior to the May 7, 2004, Advisory Action.

V. SUMMARY OF INVENTION

The present invention relates to a loudspeaker unit for improving the regenerative tone quality and, more particularly, to a loudspeaker unit particularly adapted to its environment. (Specification at 1.) The present system provides a loudspeaker unit which corrects the sound lag and phase shift attributable to the reverberation and echo of the sound in addition to the frequency characteristics of the sound produced by the system. (See specification at 2.) The present system includes a processor which compares a feedback signal input from a microphone with a sound source signal. It is important to note that claimed invention compares the output of the microphone and the actual original sound source that is driving the loudspeaker. The processing module applies correction data to the sound source signal to produce a correction signal which is then applied to an amplifier. (See specification at 5.) The intensity of the reverberation that changes the frequency characteristics are corrected according to results obtained with respect to the data of the sound source. After analyzing the frequency characteristic delay

attributable to the reverberation and the echo, the value for correction is changed to determine the correction parameter. (Specification at 8.) As such, the processor generates an output signal from the sound source using a difference in a direct output signal from the microphone with the output signal from the sound source with reference to a frequency characteristic and an echo characteristic of the sound generated from the loudspeaker, or a reverberation characteristic of the sound, including a delay time for the echo characteristics or the reverberation characteristics. (Specification at 4, et seq.)

VI. ISSUES

For the Group I claims:

Are the Group I claims patentable over U.S. Patent No. 5,172,417 ("Iwamura")?

VII. GROUPING OF CLAIMS

For purposes of this appeal brief only, and without conceding the teachings of any prior art reference, the claims have been grouped as indicated below:

Group	Claim(s)
I.	1, 4, 6, 8, and 10

In Section VIII below, Applicant has included arguments supporting the patentability of Group I as required by M.P.E.P. § 1206.

VIII. ARGUMENTS

I. Are the Group I claims patentable over Iwamura?

In the Final Office Action dated January 29, 2004, the Examiner issued a final rejection of claims 1, 4, 6, 8, and 10 under 35 U.S.C. § 102(b) as being anticipated by Iwamura. However, the Iwamura reference fails to disclose Applicant's claimed invention. Claims 2, 3, 5, 7, and 9 were objected to as being dependent from a rejected claim. The objected to claims are not addressed herein. Applicant will take appropriate steps with respect to the objected to claims after resolution of the rejected claims.

A. Anticipation

With regard to the Section 102 rejection, Appellant's submit that this rejection is improper because the Examiner failed to establish a *prima facie* case of anticipation based upon the Iwamura reference.

To anticipate a claim under 35 U.S.C. § 102, the cited reference must disclose every element of the claim, as arranged in the claim, and in sufficient detail to enable one skilled in the art to make and use the anticipated subject matter. See PPG Industries, Inc. v. Guardian Industries Corp., 75 F.3d 1558, 1566 (Fed. Cir. 1996); C.R. Bard, Inc. v. M3 Sys., Inc., 157 F.3d 1340, 1349 (Fed. Cir. 1998); MPEP § 2131, citing Verdegaal Bros. v. Union Oil Co. of California, 814 F.2d

628, 631, 2 USPQ2d 1051, 1053 (Fed. Cir. 1987)("A claim is anticipated only if each and every element as set forth in the claim is found, either expressly or inherently described, in a single prior art reference."). A reference that does not expressly disclose all of the elements of a claimed invention cannot anticipate unless all of the undisclosed elements are inherently present in the reference. See Continental Can Co. USA v. Monsanto Co., 942 F.2d 1264, 1268 (Fed. Cir. 1991). "The identical invention must be shown in as complete detail as is contained in the . . . claim." Richardson v. Suzuki Motor Co., 868 F.2d 1226, 1236, 9 USPQ2d 1913, 1920 (Fed. Cir. 1989).

Claim 1 explicitly recites

> a processor for comparing in real time a direct output signal from the microphone with an output signal from a sound source with reference to a frequency characteristic and an echo characteristic of the sound regenerated from the loudspeaker, or a reverberation characteristic of the sound, including the delay time for the echo characteristic or the reverberation characteristic, and correcting a signal from the sound source using the difference in output signal between the microphone and the sound source by reference to the frequency characteristic and the echo characteristic or the reverberation characteristic. (emphasis added)

Similarly, claim 10 recites:

> a processor for generating a processor output by correcting an output signal from the sound source using a difference in a direct output signal from the microphone with an output signal from the sound source with reference to a frequency characteristic and an echo characteristic of the sound regenerated from the loudspeaker, or a reverberation characteristic of the sound, including a delay time for the echo characteristic or the reverberation characteristic (emphasis added)

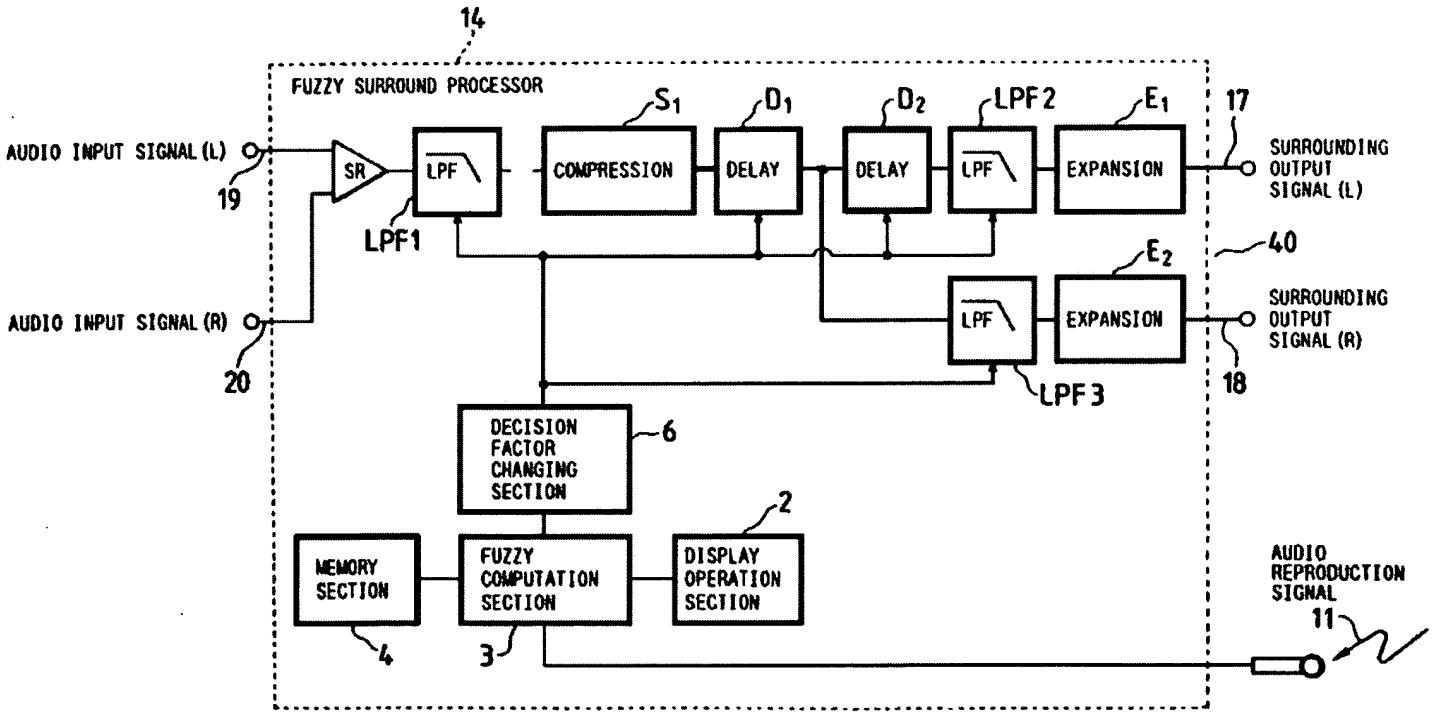
Applicant's claimed invention compares the output of the microphone and the actual original sound source that is driving the loudspeaker. Applicant's claimed invention is thus comparing the actual output of the loudspeaker to a completely different signal than that used in Iwamura. Further, Iwamura fails to implicitly or inherently disclose the signal comparison in real-time.

As illustrated in Figure 14, reproduced below, Iwamura discloses a an apparatus for controlling acoustical transfer characteristics including

> fuzzy surrounding processor 14 [that] takes in the audio input signal 11, measures the reverberation characteristics and performs fuzzy computation by using the actual reverberation characteristics and the desired reverberation characteristics as decision factors to change the characteristics of the low pass filters LPF1, LPF2, LPF3 and the delay circuits D1, D2 thereby obtaining the desired reverberation characteristics and effectively improving the presence based on the volume of the reproduced sound.

(col. 10, lns. 52-62) (emphasis added). Iwamura measures the reverberation of the sound from the loudspeaker (11) and compares it to the desired reverberation that is stored in memory 4 and adjusts the decision factors in order to obtain the desired reverberation characteristics. The decision factors are changed in element 6. Thus, Iwamura compares an output of the loudspeaker to some predefined characteristic stored in memory, not the source signal as explicitly recited in Applicant's claims.

FIG. 14



In Iwamura, the original sound source is designated as signals 19 and 20. These signals are never compared to the output of the microphone 11 as required by claims 1 and 10. Thus, Iwamura fails to disclose comparing the output of the sound source and the output of the microphone as recited in Applicant's claim.

Based on the disclosure in Iwamura the speaker output is always compared to a characteristic that is stored in memory. Because there are an almost infinite amount of characteristic patterns for fuzzy interference based on the decision factors, the Iwamura system does not store each and every predefined characteristic in memory. In fact, Iwamura states that there is no need to store all supposed characteristic patterns in a memory. The process is thereby simplified because only select characteristic patterns are stored. However, at no time is the direct output signal from the microphone compared with an output signal from the sound source with reference to a frequency characteristic and echo characteristic in real time.

In the Advisory Action, the Examiner asserts that the Iwamura reference indicates that data from the memory is not used during a real time control of the acoustic system, citing col. 11, lns. 1-13. The Examiner further asserts that the claim language "does not restrict the use of the source characteristic being stored in a memory for correcting the loud speaker audio out-put characteristics." Office Action at 4. Applicant disagrees with the Examiner's assertions.

The cited portion of Iwamura states "there is no need for storing, in a memory or the like, all supposed characteristic patterns for fuzzy interference based on given decision factors, and the computation process is simplified, thereby enabling real time adjustment." (Col. 11, lns. 9-13) (emphasis added). This cited portion of Iwamura does not state that there are no characteristic patterns stored in memory as asserted by the Examiner, merely that each and every possible characteristic pattern is not stored. Thus, when the computation process is performed, the computation utilizes a characteristic pattern that is stored in a

memory but, not each and every possible characteristic pattern is stored. Therefore, in contrast with the Examiner's assertion, Iwamura's computations always use memory.

With respect to the Examiner's assertion that the claim language does not restrict the use of the source characteristic being stored in memory, Applicants assert that, as explicitly recited in the claims, the processor generates a difference by comparing, a direct output from the microphone with an output signal from the sound source. As such, the claims are limited to an apparatus whereby the source characteristics are not stored in a memory, i.e., real-time signals.

Independent claims 1 and 10 each recite limitations not anticipated by the cited references. Because each of the independent claims are patentable, the Office should also allow their dependent claims as well.

For the reasons discussed above, the Examiner has failed to set forth a prima facie case of anticipation in the present application. As such, Applicant respectfully submits that the pending claims are not anticipated in light of the cited reference and are in condition for immediate allowance.

IX. CLAIMS INVOLVED IN THE APPEAL

A copy of the claims involved in the present appeal is attached hereto as Appendix A.

Dated: July 28, 2004

Respectfully submitted,

By_____
Ian R. Blum
Registration No.: 42,336
DICKSTEIN SHAPIRO MORIN & OSHINSKY LLP
1177 Avenue of the Americas
41st Floor
New York, New York 10036-2714
(212) 835-1400
Attorney for Applicant

APPENDIX A

Claims Involved in the Appeal of Application Serial No. 09/039,072

1. (Previously Presented) A loudspeaker unit for a sound source, the loudspeaker unit being adaptable to changing environments, the loudspeaker unit comprising:

a loudspeaker;

a microphone for picking up sound regenerated from the loudspeaker;

a processor for generating a difference by comparing, in real time, a direct output signal from the microphone with an output signal from the sound source with reference to a frequency characteristic and an echo characteristic of the sound regenerated from the loudspeaker, or a reverberation characteristic of the sound, including the a delay time for the echo characteristic or the reverberation characteristic, the processor further generating a processor output by correcting a the output signal from the sound source using the difference; and

an amplifier for amplifying the processor output.

2. (Previously Presented) A loudspeaker unit adapted to the environment according to Claim 1 wherein said processor for correcting the signal from said sound source comprising:

a first A/D converter for performing digital conversion of a sound signal outputted from the sound source;

a memory for storing a converted voice data of samples taken within a fixed time determined as a subject time for the delay of the reverberation characteristic or the echo characteristic;

a second A/D converter for performing digital conversion of the feedback signal outputted from said microphone as the feedback data;

a successive comparison analysis part for successively comparing said feedback data with the stored voice data, analyzing the intensity of the reverberation characteristic or the echo characteristic and outputting the result as a correction parameter;

a regenerative signal processing part for adding data corrected by said correction parameter to the stored voice data and processing the result as the regenerative data; and

a D/A converter for converting said regenerative data to an analog signal.

3. (Previously Presented) A loudspeaker unit adapted to the environment according to Claim 1 wherein a successive comparison analysis part performs processing by adding antiphase feedback data to voice data so that the difference between said voice data obtained as the serial data and said feedback data becomes a fixed value or 0.

4. (Previously Presented) A loudspeaker unit adapted to the environment according to Claim 1 wherein,

the frequency comparison of the characteristic and the comparison of the characteristic of the echo or the reverberation each including the delay time are learned by arithmetic and a signal to be sent to the loudspeaker is corrected according to the learned result.

5. (Previously Presented) A loudspeaker unit adapted to the environment according to Claim 2, wherein, the frequency comparison of the characteristic and the comparison of the characteristic of the echo or the reverberation each including the delay time are learned by arithmetic and a signal to be sent to the loudspeaker is corrected according to the learned result.

6. (Previously Presented) A loudspeaker unit adapted to the environment according to Claim 1 wherein,

the frequency comparison of the characteristic and the comparison of the characteristic of the echo or the reverberation each including the delay time are intermittently performed and a signal to be sent to the loudspeaker is corrected according to the comparison result.

7. (Previously Presented) A loudspeaker unit adapted to the environment according to claim 2, wherein, the frequency comparison of the characteristic and the comparison of the characteristic of the echo or the reverberation each including the delay time are intermittently performed and a signal to be sent to the loudspeaker is corrected according to the comparison result.

8. (Previously Presented) A loudspeaker unit adapted to the environment according to Claim 4 wherein,

the frequency comparison of the characteristic and the comparison of the characteristic of the echo or the reverberation each including the delay time are intermittently performed and a signal to be sent to the loudspeaker is corrected according to the comparison result.

9. (Previously Presented) A loudspeaker unit adapted to the environment according to Claim 5 wherein,

the frequency comparison of the characteristic and the comparison of the characteristic of the echo or the reverberation each including the delay time are intermittently performed and a signal to be sent to the loudspeaker is corrected according to the comparison result.

10. (Previously Presented) A loudspeaker unit for a sound source, the loudspeaker unit being adaptable to changing environments, the loudspeaker unit comprising:

a loudspeaker;

a microphone for picking up sound regenerated from the loudspeaker;

a processor for generating a processor output by correcting an output signal from the sound source using a difference in a direct output signal from the microphone with an output signal from the sound source with reference to a frequency characteristic and an echo characteristic of the sound regenerated from the loudspeaker, or a reverberation characteristic of the sound, including a delay time for the echo characteristic or the reverberation characteristic; and

an amplifier for amplifying the processor output.